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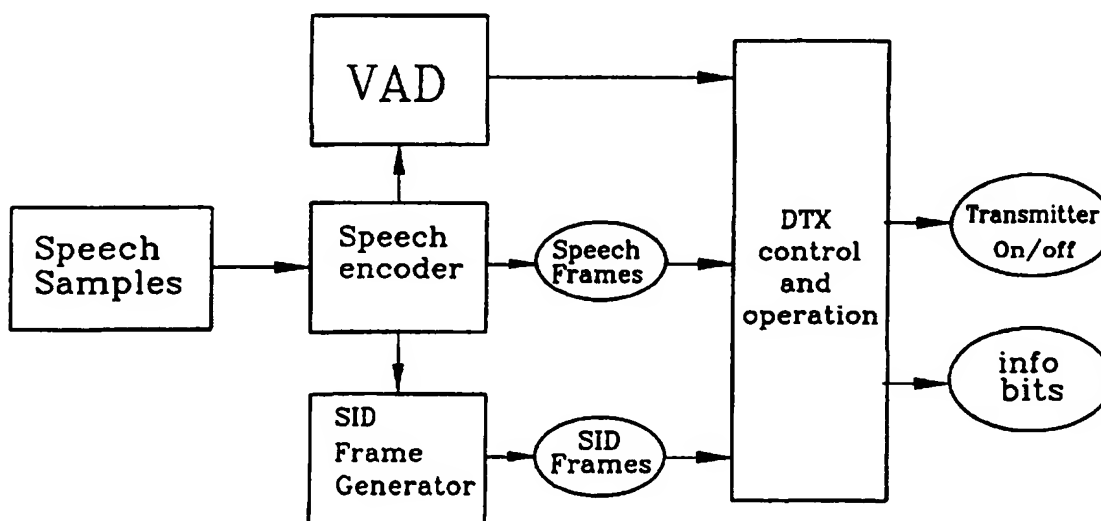
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(54) Title: ARRANGEMENT AND METHOD RELATING TO SPEECH TRANSMISSION AND A TELECOMMUNICATIONS SYSTEM COMPRISING SUCH ARRANGEMENT



## (57) Abstract

The present invention relates to an arrangement and a method in a speech transmission system particularly with a speech frame structure. Means (VAD) are provided for detecting if a signal comprises speech information and detecting means are provided for detecting the presence of a bad or lost frame. When it has been detected that a speech frame has been corrupted or lost during transmission, it is replaced by a frame representing mainly background noise or a combination of at least one such frame and at least one correctly received speech frame.

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## Title:

5 ARRANGEMENT AND METHOD RELATING TO SPEECH TRANSMISSION AND  
A TELECOMMUNICATIONS SYSTEM COMPRISING SUCH ARRANGEMENT

## FIELD OF THE INVENTION

10 The present invention relates to an arrangement and a method  
relating to speech transmission wherein the transmitted  
signals are divided into a frame structure. The invention  
also relates to a telecommunications system comprising an  
arrangement relating to speech transmission.

15

## STATE OF THE ART

In digital telecommunications systems a frame structure is  
almost always used and speech is transmitted in speech  
(traffic) frames. A frame here relates to an information  
20 block comprising a given number of digital information bits.  
When speech is to be transmitted the solution is not  
straightforward since on one hand both speech and background  
noise, which may vary to a great extent, is present and on  
the other hand a human speaker normally does not speak  
25 uninterruptedly but now and then makes pauses and remains  
silent. Furthermore, frames or speech-frames may be bad,  
i.e. lost or corrupted during transmission.

When a transmitted frame is bad or lost it will generally be  
30 replaced since normal decoding of such frames would produce  
noise effects which are very annoying for a listener.

GSM Recommendations GSM 06.11, October 1992, "Substitution  
and Muting of Lost Frames for Full-Rate Speech Channels"  
35 relates to muting when the full-rate speech coding is  
applied, i.e. they define a frame substitution and muting

procedure to be used by the receiving side when one or more lost speech frames or SID frames are received.

5 When speech frames have been lost, the speech volume is decreased. A muting technique is disclosed through which the output level is decreased gradually resulting in silencing of the output after a maximum 320 ms. This means that silence will be received after max 320 ms which can be very annoying since it is an abrupt change from speech plus  
10 background noise to silence. Often a period which is shorter than 320 ms is used in practice which can be even more annoying.

15 If aural information comprises both speech and background noise mixed, muting towards silence induces inconvenient sparkling. Thus, for a number of known muting algorithms which are applied on disturbed speech coding parameters, the background noise chops down to silence and this may happen more than once a second. Furthermore, known solutions do not  
20 take into account such situations when background noise is present such as babble, car-noises etc., which however are realistic traffic cases.

#### SUMMARY OF THE INVENTION

25 A problem in speech transmission is that the sound (aural) information may comprise speech or background noise or speech and background noise mixed. In the last case, and if muting towards silence, in the case of frames being lost or corrupted during transmission, inconvenient sparkling is  
30 induced. The reason for this is the alternation between complete silence and speech or noise.

It is an object of the present invention to provide an arrangement and a method respectively in a speech  
35 transmission system wherein discomforting effects because of speech frames being lost or corrupted during transmission are reduced to a minimum.

Particularly it is an object of the invention to provide an arrangement and a method respectively through which discomforting effects can be minimized or avoided when two or more consecutive speech frames are lost.

5

It is another object of the present invention to provide an arrangement and a method respectively which can be applied regardless of whether the transmission is discontinuous or continuous.

10

Generally it is an object of the invention to provide an arrangement and a method respectively which is flexible, which can be applied in different systems having different requirements as to power savings etc. and which is reliable, efficient and which can easily be applied.

15

It is also an object of the present invention to provide a telecommunications systems comprising an arrangement in a speech transmission system which meets the abovementioned objects.

20

These as well as other objects are achieved through an arrangement and a method respectively wherein if a frame is lost or corrupted during transmission, it can be replaced by a frame representing mainly background noise. Alternatively it is replaced by a combination of at least one frame representing mainly background noise and at least one correctly received speech frame. If particularly two or more consecutive frames are corrupted or lost during transmission, they are replaced by frames which are combinations of background noise frames and speech frames in such a way as to gradually approach background noise.

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At least one background noise frame must in some way be available on the receiving side. In a particular embodiment the DTX-function (described in GSM recommendations GSM 06.31 "Discontinuous Transmission (DTX) for full-rate Speech

Traffic Channels") is applied and SID frames provided by the DTX function generated at the transmitting end are used.

5 In another embodiment SID frames are generated at the transmitting end and transmitted during periods of no speech although DTX is not used. In still another embodiment frames representing background noise (e.g. SID frames) are generated at the receiving side. In another alternative  
10 embodiment, a default SID frame is used on the receiving side, which is used when DTX is not activated or not used.

Generation of noise as such can be done in different ways and it is supposed to be known.

15 Also the bad frame indicating means can be any adequate bad frame indicating means.

In a particular embodiment of the invention is dealt with the problem when occasionally frames which are not bad are  
20 received in periods when bad frames dominate. A change frame comfort noise to full volume speech frames may then be disturbing.

According to the invention may therefore, if a speech frame  
25 is correctly received and the at least two preceding speech frames were lost or corrupted during transmission, the correctly received speech frame be replaced by a frame which is a combination of the correctly received speech frame and at least one frame representing background noise.  
30 Particularly, if a given number of consecutive correctly received frames are preceded by a given number of bad frames, the correctly received frames are replaced by frames which are combinations of speech frames and background noise frames so as to gradually approach speech.

35

The invention thus proposes solutions in which ramping down is provided or ramping down and ramping up or just ramping up.

5 For the latter case an arrangement in a speech transmission is given wherein signal are divided into a frame structure, comprising means for detecting if a signal contains speech information and means for detecting if frames are bad or not. If a speech frame is correctly received, it is examined  
10 if a given number of frames directly preceding the received frame are bad, and if so, the correctly received speech frame is replaced by a frame representing a combination of background-noise and a correctly received speech frame.

15 Particularly, if a given number of consecutive non-bad frames are preceded by a given number of bad frames, the non-bad frames are replaced by frames which are combinations of speech frames and background noise frames so as to gradually approach speech.

20 Particular embodiments of the invention relate to the GSM system. For these embodiments the GSM recommendations as referred to in the application are applicable and define a number of functions etc.

25 When discussing a receiving and a transmitting side respectively, for example in a mobile communication system, it may relate to e.g. a radio base station both as a sender sending to a mobile station (a downlink connection) and to  
30 a radio base station as a receiving arrangement whereas a mobile station is the sending arrangement (an uplink connection).

It is an advantage of the invention that if frames are lost  
35 or corrupted during transmission, the effects thereof are reduced considerably as compared to hitherto known systems.

The great flexibility in the applicability of the invention is also a great advantage and it can be used in generally every digital telecommunications system for speech transmission. The invention is mainly focused on digital,  
5 frame structure based, systems as referred to in the state of the art.

The invention can though be applied in analog system; this however requires additional installations as will be referred to in the detailed description of the invention.  
10

#### BRIEF DESCRIPTION OF THE DRAWINGS

The invention will in the following be further described in a non-limiting way under reference to the accompanying  
15 drawings wherein:

- Fig. 1 is a block diagram illustrating the transmitting side in a first embodiment of the invention,
- 20 Fig. 2 is a block diagram of the receiving side corresponding to the embodiment of Fig. 1,
- Fig. 3 illustrates a flow diagram of the muting according to the invention,
- 25 Fig. 4 illustrates a table describing the muting procedure in detail,
- Fig. 5 shows a further embodiment of the invention in which SID-frames are assumed not to be transmitted and  
30
- Fig. 6 illustrates application of the invention on an analog system
- 35 Fig. 7 shows a flow diagram as in Fig. 3 relating to an alternative embodiment comprising ramping up and

Fig. 8 shows an alternative embodiment also comprising ramping up.

#### DETAILED DESCRIPTION OF THE INVENTION

5 The invention will first be further described in relation to the full rate speech coder of the GSM system although the invention by no means is limited to said system. In an alternative embodiment (not further described) half-rate speech transcoding on half-rate speech channels is applied.

10 In the cellular mobile system GSM speech is transmitted in the form of speech frames comprising encoded speech data as referred to earlier in the application. The arrangement comprises means for detecting if voice activity is present or not, i.e. frames containing speech are distinguished from

15 frames containing silence or just background noise. These voice activity detecting means are generally referred to as a voice activity detector VAD. The VAD algorithm is defined in the GSM Recommendations GSM 06.32, "Voice Activity Detection".

20 In the following a first embodiment will be discussed in relation to Fig. 1 relating to the GSM system operating in discontinuous transmission mode which is defined in the GSM Recommendations GSM 06.31 "Discontinuous Transmission (DTX)

25 for Full-Rate Speech Traffic Channels". Discontinuous transmission DTX is a mechanism which allows a radio transmitter to be switched off most of the time when there is no speech, i.e. during speech pauses. Two reasons for doing so is to save power and to reduce the over-all

30 interference level on the air. Then background noise is estimated by an algorithm, through averaging speech parameters in four consecutive speech frames, a voice activity detector (VAD) as referred to above determines whether an incoming signal contains speech information or

35 not.

In periods when the VAD indicates no speech, a SID frame is sent with regular intervals. In the periods between these updates the transmitter can be turned off.

- 5 The GSM system discloses a full-rate speech coding algorithm which performs a compression of incoming speech samples reducing the bitrate with approximately 90%. The GSM full-rate speech coding is discussed in GSM Recommendations 06.10, January 1990, "GSM Full-Rate Speech Transcoding".
- 10 However, using this generally makes the speech channel becoming less robust to induced bit errors.

Fig. 1 shows the transmitting side. Incoming speech samples are speech encoded to reduce the bitrate. The output from

15 the speech encoder is a given number of speech frames every second.

The voice activity detector has an output signal VAD-flag, that indicates if the present frame contains speech

20 information or not.

When a number of consecutive frames containing no speech information has been detected, a SID frame generator calculates a SID frame based on the current frame and a

25 given number of old frames. In periods of no speech activity, SID-frames can, on the receiver side, be used to generate background noise over a longer period of time than an ordinary speech frame.

30 Through the SID frame generator SFG the characteristics of the background noise are measured in case of no speech and a SID frame (containing parameters describing background noise) is produced.

35 The DTX control and operation has two output signals. Info bits are normally the speech frames from the speech encoder, and the "transmitter on" flag is set true.

In case of several speech frames marked with "no VAD", at least as many as required to produce a SID frame based on just "no VAD" marked frames, the info bits are set to be the SID frame.

5

In periods where the info bits are set to be SID-frames, the "transmitter on" flag is set to false, except for some regular updates.

10

Figure 2 shows the receiving side. The first input signal comprises the info bits, received from a non-perfect channel. The second is the BFI (Bad Frame Indication) flag from a channel decoding or equalizing device marking bad frames. A frame can be marked as bad for two reasons, namely that some info bits are suspected to be erroneous, or that no frame is received, possible because the transmitter has been turned off.

15

20

It should be noted however that the present invention only relates to frames bad in the sense that they are lost or corrupted during transmission. The invention is thus not concerned with deliberate transmission pauses due to DTX.

25

The DTX control and operation unit determines if the received info bits comprise a SID frame or a speech frame.

In case of a speech frame, it is speech decoded, producing speech samples. In case of a SID frame, the comfort noise generator generates a frame that describes background noise.

30

In case of a BFI marked frame, the speech frame substitution unit produces a speech frame which is sent to the speech decoder or a SID-frame which is sent to the Comfort Noise Generator. The produced frame is in this case based on (1)

35

previously received speech frames, (2) a previously received SID-frame and (3) current received bad frame.

The basics of discontinuous transmission DTX will now be briefly discussed. The DTX function requires a VAD on the transmit side, evaluation of background noise on the transmit side for transmitting characteristic parameters to the receiving side and generation of comfort noise similar thereto on the receive side when radio transmission is cut.

This is further described in GSM Recommendations GSM 06.31. The DTX operation mode provides for having the transmitters switched on only as long as the frames comprise useful information. The DTX mechanism is implemented in the DTX handlers both on the transmit side and on the receive side and comprises a VAD on the transmit side as discussed above, a unit for evaluating the background noise on the transmit side in order to transmit characteristic parameters to the receive side and a unit for generating comfort noise on the receive side during periods when the radio transmission is cut. Through the VAD is determined whether a specific block of 20 ms from the speech coder comprises speech or not. Due to the changes both in noise level and in noise spectrum in mobile environments, the VAD generally has to be constantly adapted thereto. The VAD is an energy detector wherein the energy of a filtered signal is compared to a threshold and speech is indicated whenever the threshold is exceeded.

The insertion of comfort noise will now be briefly discussed. When a transmission is on, the background noise is transmitted together with the speech. As a speech period ends, the connection is off and the perceived noise will drop to a very low level. This would produce a step modulation of noise which would be perceived as annoying and it may also reduce the accuracy of speech if it were to be presented to a listener without any modification. This is called a noise contrast effect and this is reduced through

the insertion of an artificial noise here referred to as comfort noise at the receiving end when speech is absent. The parameters which are needed for generation of the comfort noise are sent as background noise parameters before transmission is cut off and thereafter on scheduled positions. The frames comprising this background noise are the SID-frames as referred to above. This however do not relate to frames lost/corrupted during transmission.

Speech frames may be lost or bad for various reasons. For example in the receiver frames may be lost due to transmission errors or frame stealing for the fast associated control channel FACCH. Frames may also be lost during handover. To reduce the consequences of one single lost frame, a scheme may be used according to which the lost speech frame is substituted by a predicted frame based on the previous frame. For several consecutive lost frames however muting has to be done. Advantageous ways of doing this will now be more thoroughly described.

In the embodiment illustrated in Figs 1 and 2 relating to a full-rate transcoding case, the output from the speech-coder can be a block of 260 bits every 20ms which gives a bit rate of 13kbit/s. A known coding scheme can be used e.g. as described in the GSM Recommendations 06.10. The encoded speech at the output of the speech encoder is delivered to the channel coding functions in order to produce an encoded block. As to the receiving part as illustrated in Fig 2, the corresponding inverse operations take place.

Now muting towards background noise will be more thoroughly described in relation to the muting algorithm.

Figure 3 shows a flow diagram of the muting algorithm, and the choice of output device of the speech samples. A variable "Counter of Bad Frames" (CBF) is introduced. "Mute

Period" MP is a constant which is connected to the length of the mute table shown in figure 4.

When a frame is received the BFI indicates whether it is a bad frame or not. If it is settled that it is not a bad frame, the number of bad frames which have been received as indicated by the CBF number is reset to 0 and the correctly received speech frame is delivered as output data and hence a speech frame is output. On the other hand, if BFI indicates that the frame is bad, the variable indicating the number of consecutive bad frames that have been received, CBF, is increased by 1. Then it is examined if the number of consecutive bad frames received, CBF, exceeds the length of the mute period in frames, MP. The length of the mute period MP is a given constant giving the number of frames during which muting is to be effected. If thus the number of consecutive bad frames received, CBF, exceeds the length of the mute period, MP, the preceding correctly received SID frame is used for generation of comfort-noise. Thereupon a SID frame is delivered as output data. (The mute period MP is e.g. taken to 4.) If on the other hand the number of consecutively received bad frames, CBF, is between 1 and MP, a muting algorithm is used to calculate a number of parameters to be used by the speech decoder. The parameters used by the speech decoder are for GSM defined in GSM 06.10, 06.11 and 06.12. In the exemplifying embodiment the parameters GAIN[N] and XMAX[N] are given by the muting algorithm described in Fig. 3 and 4. CBF=(1-4) is a description of how to combine the parameters from the different frames available. CBF $\geq$ 5 shows how plain SID frames are sent to the Comfort Noise Generator.

The transition from comfort noise to non-muted speech within one frame when a good frame is received, as described in figure 3, is relevant in disturbance conditions as occasional fadings or interferences.

However, under very bad conditions for radio transmission a problem occurs with receiving occasional frames that are not bad in periods where receiving BFI-marked frames is dominant. The change from comfort noise to the full volume speech frame and the muting to comfort noise again could create an disturbing transient on both the level and the spectrum.

10 In an advantageous embodiment this is dealt with as schematically illustrated in the flow diagram of Fig. 7.

When a frame is received the BFI indicates whether it is a bad frame or not. If the frame is considered as bad the same muting procedure as described above is applied. On the contrary, if BFI indicates that the frame is not bad, a check is done to see if the previous frame was speech decoded without manipulation or not, i.e. if CBF is zero or not. If CBF is equal to zero the frame is delivered to the speech decoder without any manipulation. On the other hand, if CBF is greater than zero it is examined if in the comfort noise generation state or in the muting period, i.e. if  $CBF > MP$ . If in the comfort noise state the CBF is set to MP. On the other hand, if in the muting period the CBF is decreased by one. Then the same table as disclosed in figure 4 may be re-used for the ramping up of the speech. Finally the combined speech and comfort noise parameters are passed to the speech decoder.

30 In still another embodiment the counter CBF may be limited to values up to and including  $MP + 1$ .

Ramping between speech frames and noise frames can then be done as illustrated in Fig. 8. As an example the table of fig. 4 may be used to calculate the output frames.

The GSM full rate speech coding scheme at 13 kbit/s is called RPE-LTP (Regular Pulse Excitation-Long Term Prediction).

5 The speech coder first cuts the speech, represented by 13  
bit linear PCM samples sampled at a rate of 8 kHz, into 20  
ms slices, called frames. Such a frame of 160 samples is  
then pre-processed to produce an offset-free signal, which  
is then subjected to a first order pre-emphasis filter. The  
10 resulting 160 samples are then analyzed to determine the  
coefficients for the short term analysis filter, which is  
used for modelling the overall spectral envelope. This is  
done by using LPC, Linear Prediction Coding, analysis, i.e.  
to minimise the energy of the signal obtained when filtering  
15 the 160 samples through the reverse LPC filter. These  
parameters are then used for the filtering of the same 160  
samples. The result is 160 samples of the short term  
residual signal. The filter parameters, termed reflection  
coefficients, are transformed to log area ratios, LARs,  
20 before transmission.

The short term residual signal is then divided into four  
sub-frames of 40 samples each.

25 Before the processing of each sub-block, the estimates of  
the parameters of the long term analysis filter are updated,  
based on stored reconstructed short term residual from the  
three last sub-frames together with current one. The long  
term analysis filter is determined to describe the  
30 similarity of successive periods of voiced segments. The  
parameters are denoted LTP lag and LTP gain, LTP denotes  
long term prediction. LTP lag gives an index of the  
periodicity and the LTP gain gives a value of the  
correlation energy, i.e. the similarity of the sub-blocks.

35

The LTP filter gives a prediction of the 40 short term  
residual samples of the sub-frame. Subtracted from the 40

short term residual samples, a block of 40 long term residual samples, for the sub-frame, is obtained. This is then repeated for all sub-frames.

- 5     These long term residual samples are then further compressed by RPE, regular pulse excitation, analysis. The result is a set of RPE-parameters, of which the Xmax parameter gives the estimated sub-block amplitude.

10

This just relates to one particular embodiment and of course the table can take many other forms; i.e. the output frame does not have to vary according to the pattern given here  
15     but according to any other pattern and the mute period does not have to be 4 but can also take other values.

In an advantageous embodiment, one or more frames representing background noise can be stored in the system,  
20     either permanently or temporarily. Irrespectively of whether it is stored in a mobile station or a base station or any other part of the system it can be stored therein upon the fabrication thereof or when it is programmed. It might also be stored temporarily for a call or for any desired period.

25

An operator of a network has the possibility to configure the network in such a way as to not use the discontinuous transmission DTX function. It is also possible for the network operator to leave the choice to the individual users  
30     who then can choose whether or not they want to use the DTX function.

However, when the DTX function is used, SID frames will arrive with a given regularity describing the background  
35     noise during periods of no speech. If a SID frame is valid it should be saved. The SID frame generator and the comfort noise generator which are arranged in the system to provide

DTX functionality are used to provide access to appropriate background noise on the receiving side.

Fig. 5 relates to the receiving side of a further embodiment with no DTX functionality. The received info bits will then always be speech frames. A SID frame generator is introduced, which generates SID frames based on the received speech frames. A VAD is also implemented. In case of no voice activity for a certain number of frames the SID frame from the SID Frame Generator will be stored in the Speech Frame Substitution unit for possible further use. In case of reception of a BFI-marked frame, speech frame substitution will be done according to the algorithms described in Figs. 3 and 4. Of course ramping up as described in Figs. 7 and 8 can also be applied here.

According to a further embodiment of the invention wherein reference can be made to figures 1 and 2, a system not using DTX can force SID frames in periods of no speech. The SID frames can be used on the receiving side by the Speech Frame Substitution Unit. According to one particular embodiment these SID frames can be sent e.g. once a second if VAD indicates no speech for a given number of frames. They can be calculated in a number of different ways.

This modification will not induce any noticeable change for the user when the channel conditions are good. Furthermore the "forced" SID-frames are just stuffed in between speech frames in periods when no speech activity is detected.

The receiving side saves the last accepted (not BFI-marked) SID frame for use when needed. In case of reception of a BFI-marked frame, speech frame substitution will be done according to the algorithms described in Figs. 3 and 4. Also here ramping up can be provided as described earlier.

Fig. 6 illustrates a further embodiment showing how the inventive concept of the present invention can be applied in an analog system. The analog speech signal is first sampled in an A/D-device, and then after the bad speech concealment measure returned to analog. This whole unit can be implemented on the receiving side. In this case no BFI is available. Necessary for operation is thus a "Bad Channel Indication" (BCI) signal which indicates (to an arrangement 10 which can be of the kind as illustrated in Fig. 5) in which periods the received analog signal is bad.

## CLAIMS

- 5      1. Arrangement in a speech transmission system, wherein  
signals are divided into a frame structure, comprising means  
for detecting if a signal contains speech information and  
means for detecting if a frame has been corrupted or lost  
during transmission,  
10      c h a r a c t e r i z e d   i n ,  
that if a speech frame is corrupted or lost during  
transmission it is replaced by a frame representing mainly  
background noise or a combination of at least one such frame  
and at least one correctly received speech frame.
- 15      2. Arrangement according to claim 1,  
c h a r a c t e r i z e d   i n ,  
that if at least two consecutive frames are corrupted or  
lost during transmission, those frames are replaced by  
20      frames which are combinations of background noise frames and  
speech frames in such a way as to gradually approach  
background noise.
- 25      3. Arrangement according to claim 1 or 2,  
c h a r a c t e r i z e d   i n ,  
that the speech transmission system uses discontinuous  
transmission.
- 30      4. Arrangement according to claim 1, 2 or 3,  
c h a r a c t e r i z e d   i n ,  
that frames representing background noise (SID frames) are  
generated at the transmitting end during speech pauses and  
used in the replacement procedure at the receiving end.
- 35      5. Arrangement according to claim 1, 2 or 3,  
c h a r a c t e r i z e d   i n ,

that frames representing background noise are generated at the receiving end.

6. Arrangement according to claim 1, 2 or 3,  
5 c h a r a c t e r i z e d i n ,  
that at least one frame representing background noise is temporarily or permanently stored in the system.
7. Arrangement according to any of the preceding claims,  
10 c h a r a c t e r i z e d i n ,  
that if a speech frame is correctly received and the at least two preceding speech frames were lost or corrupted during transmission, the correctly received speech frame is replaced by a frame which is a combination of the correctly  
15 received speech frame and at least one frame representing background noise.
8. Arrangement according to claim 7,  
c h a r a c t e r i z e d i n ,  
20 that if a given number of consecutive correctly received frames are preceded by a given number of bad frames, the correctly received frames are replaced by frames which are combinations of speech frames and background noise frames so as to gradually approach speech.
- 25 9. Arrangement according to anyone of claims 1 to 6,  
c h a r a c t e r i z e d i n ,  
that if a number of correctly received speech frames follow after a number of badly received speech frames, the first  
30 correctly received speech frames are replaced by frames which are combinations of correctly received speech frames and at least one frame representing background noise.
- 35 10. Arrangement according to claim 9,  
c h a r a c t e r i z e d i n ,  
that the output frames gradually approach pure speech frames.

11. Arrangement in a speech transmission system wherein signals are divided into a frame structure, comprising means  
5 for detecting if a signal contains speech information and means for detecting if frames are bad or not,  
c h a r a c t e r i z e d i n ,  
that if a speech frame is correctly received, it is examined  
if a given number of frames directly preceding the received  
10 frame are bad, and if so, the correctly received speech frame is replaced by a frame representing a combination of background-noise and a correctly received speech frame.

12. Arrangement according to claim 11,  
15 c h a r a c t e r i z e d i n ,  
that if a given number of consecutive non-bad frames are preceded by a given number of bad frames, the non-bad frames are replaced by frames which are combinations of speech frames and background noise frames so as to gradually  
20 approach speech.

13. Telecommunications system comprising a number of receiving arrangements and a number of transmitting arrangements wherein audio signals divided into frames of  
25 encoded data are transmitted between transmitting and receiving arrangements and wherein the system comprises encoding means and decoding means, audio detecting means (VAD) for detecting if speech activity is present in transmitted signals, means for indicating bad frames (BFI)  
30 and noise generating means,  
c h a r a c t e r i z e d i n ,  
that if the bad frame indicating means (BFI) detects that a speech frame is lost or corrupted during transmission, it is replaced by a frame representing mainly background noise or  
35 a combination of at least one such frame and at least one correctly received speech frame.

14. Telecommunications system according to claim 13,  
c h a r a c t e r i z e d i n,  
that if at least two consecutive frames are corrupted or  
lost during transmission those frames are replaced by frames  
which are combinations of background noise frames and speech  
frames in such a way as to gradually approach background  
noise.
15. Method for improving speech quality in a speech  
transmission system wherein the speech signals are divided  
into a frame structure, comprising the steps of:  
- detecting if a speech frame has been lost or corrupted  
during transmission and  
- replacing a lost or corrupted frame by a frame  
representing mainly background noise or at least one such  
frame in combination with at least one correctly received  
speech frame.
16. Method according to claim 15,  
c h a r a c t e r i z e d i n,  
that if at least two consecutive frames are corrupted or  
lost during transmission those frames are replaced by frames  
which are combinations of background noise frames and speech  
frames in such a way as to gradually approach background  
noise.

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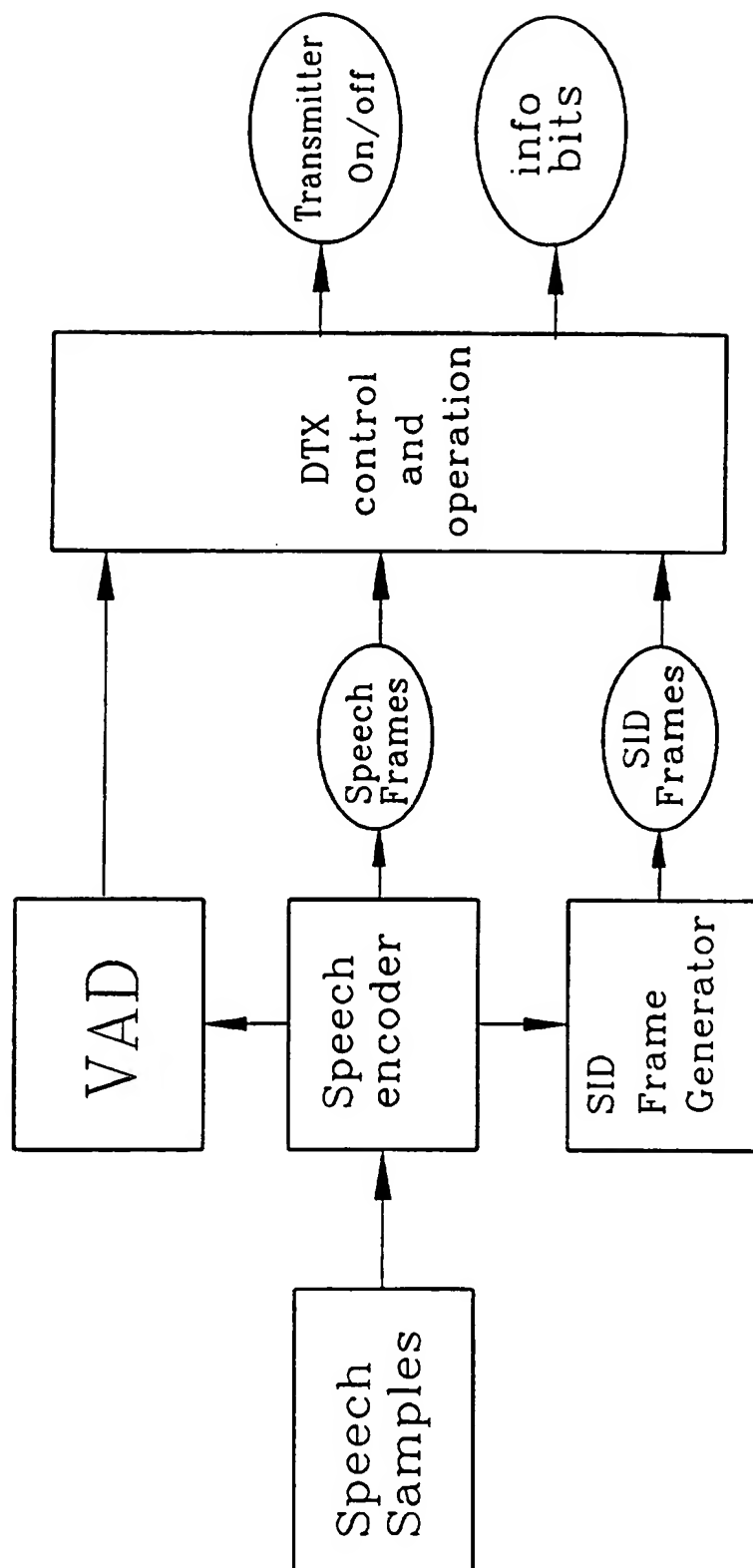


FIG. 1

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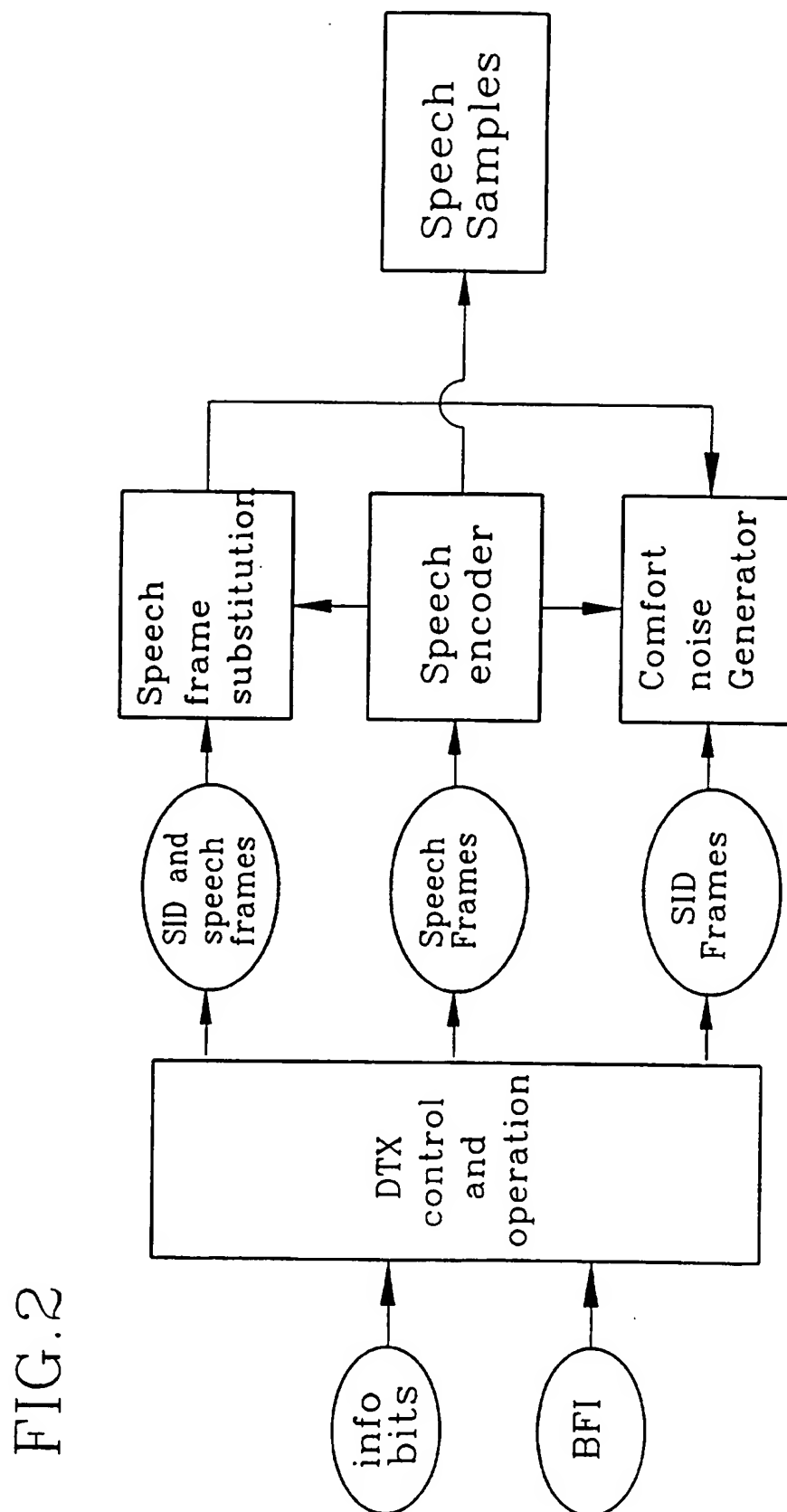


FIG. 3

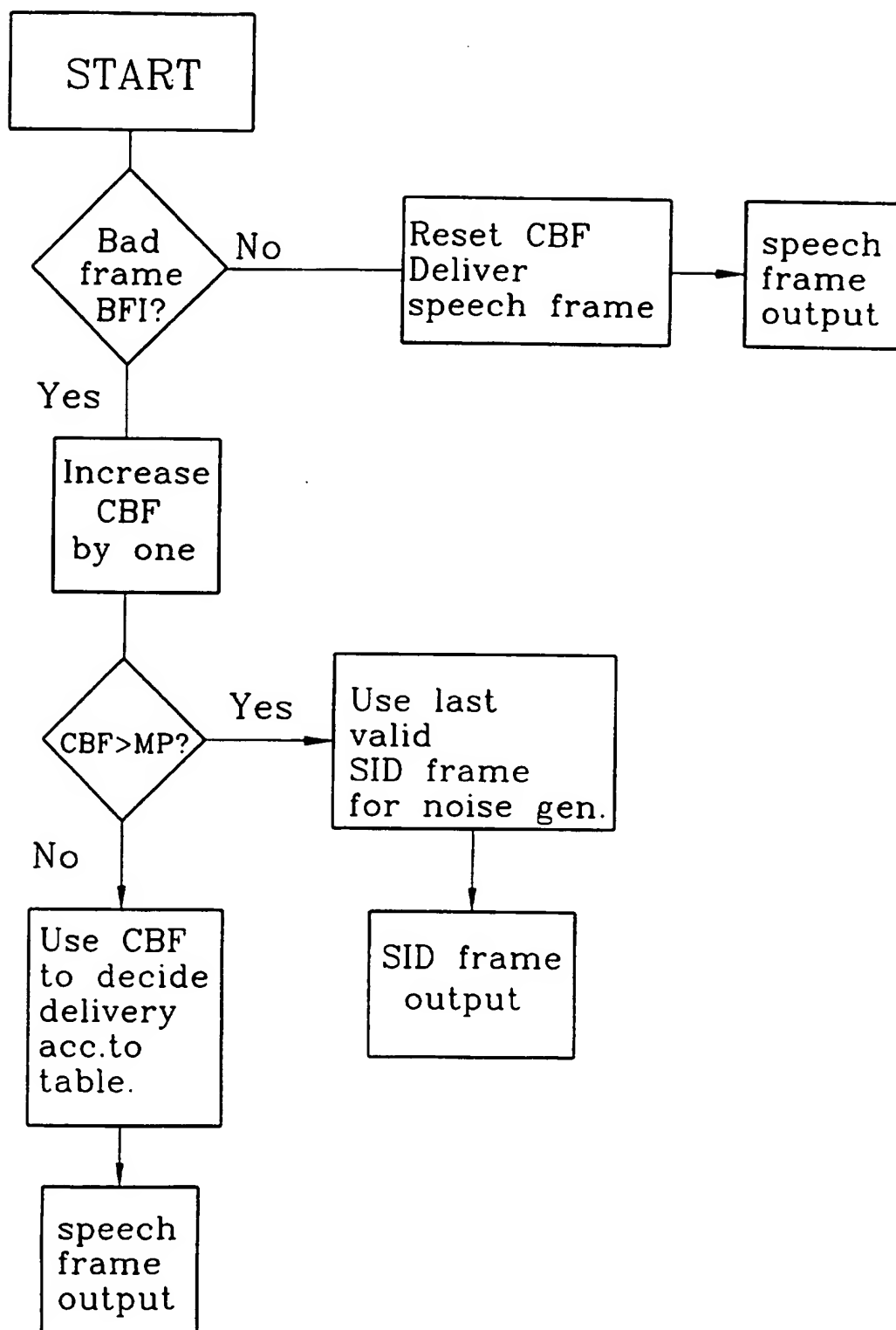


FIG. 4

CBF	OUTPUT
0	Not bad. Use received frame.
1	Repeat all parameters of last correctly received frame
2	<ul style="list-style-type: none"> <li>* Repeat all LAR parameters and the LTP lag from last correctly received frame.</li> <li>* Mute the LTP gain and Xmax parameters like: 75% from last correctly received speech frame + 25% from last correctly received SID frame.</li> <li>* Use the other parameters from current received frame.</li> </ul>
3	<ul style="list-style-type: none"> <li>* Repeat all LAR parameters and the LTP lag from last correctly received frame.</li> <li>* Mute the LTP gain and Xmax parameters like: 50% from last correctly received speech frame + 50% from last correctly received SID frame.</li> <li>* Use the other parameters from current received frame.</li> </ul>
4	<ul style="list-style-type: none"> <li>* Repeat all LAR parameters and the LTP lag from last correctly received frame.</li> <li>* Mute the LTP gain and Xmax parameters like: 25% from last correctly received speech frame + 75% from last correctly received SID frame.</li> <li>* Use the other parameters from current received frame.</li> </ul>
≥ 5	Repeat last correctly received SID-frame.

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FIG. 5

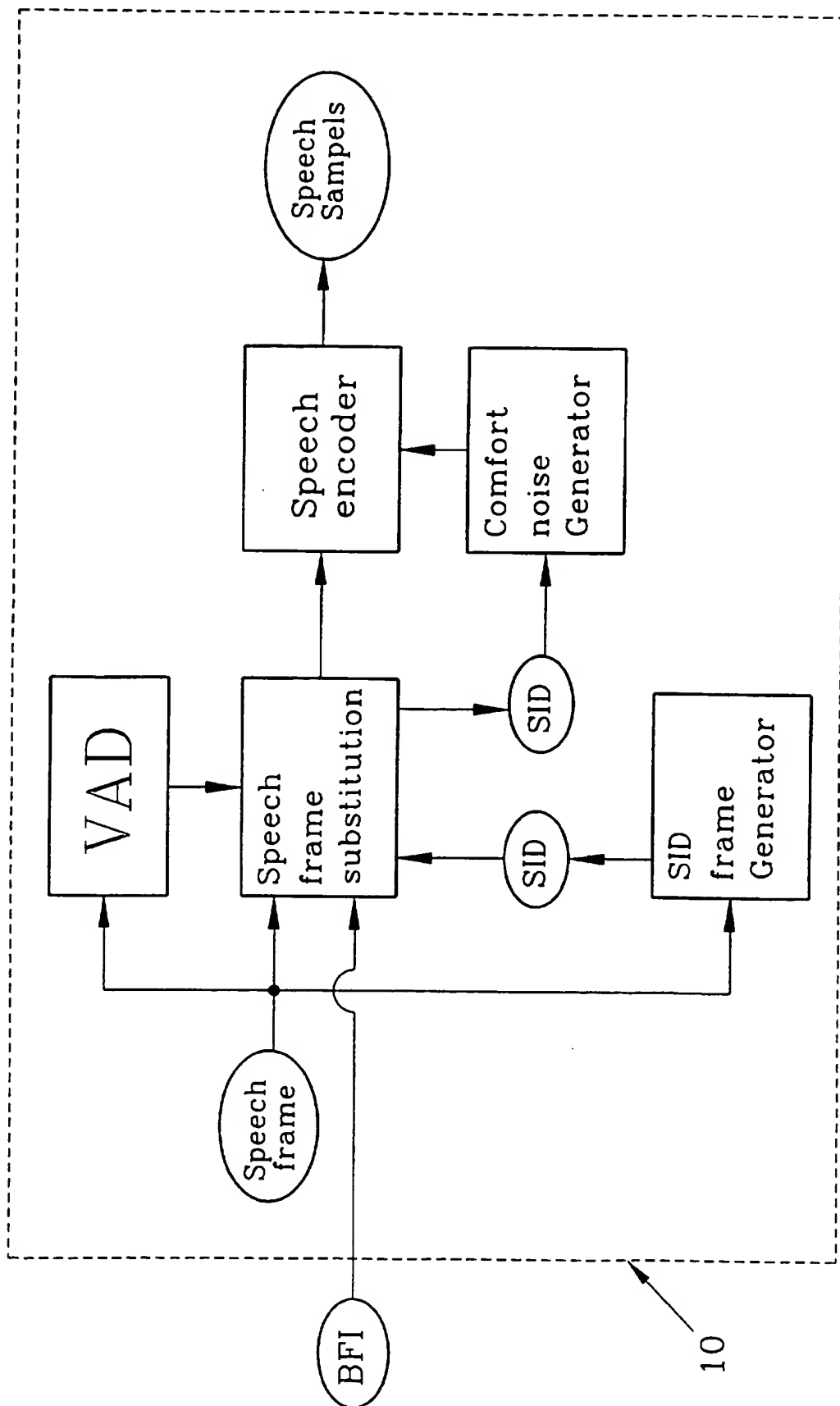
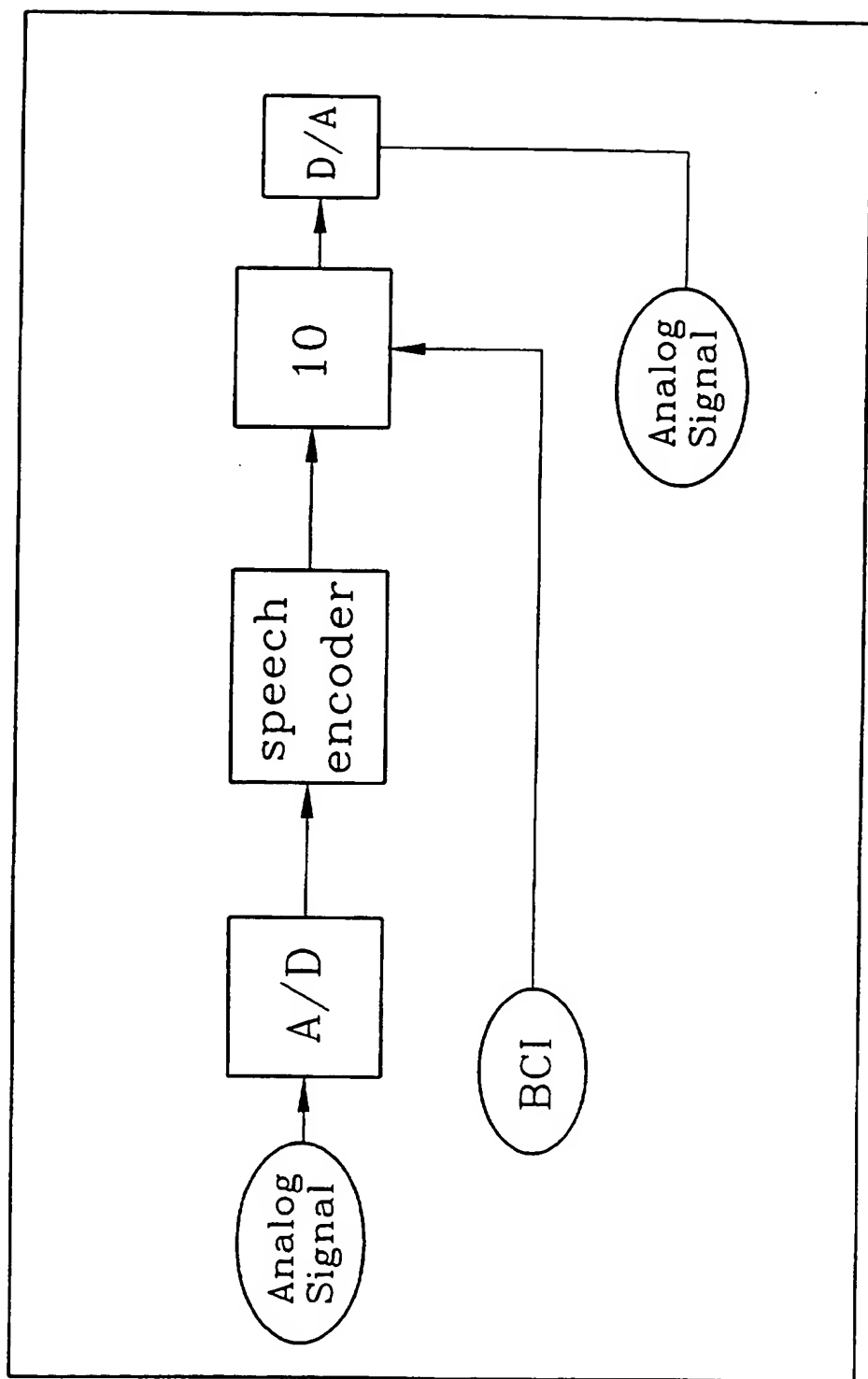


FIG. 6

Receiver



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FIG. 7

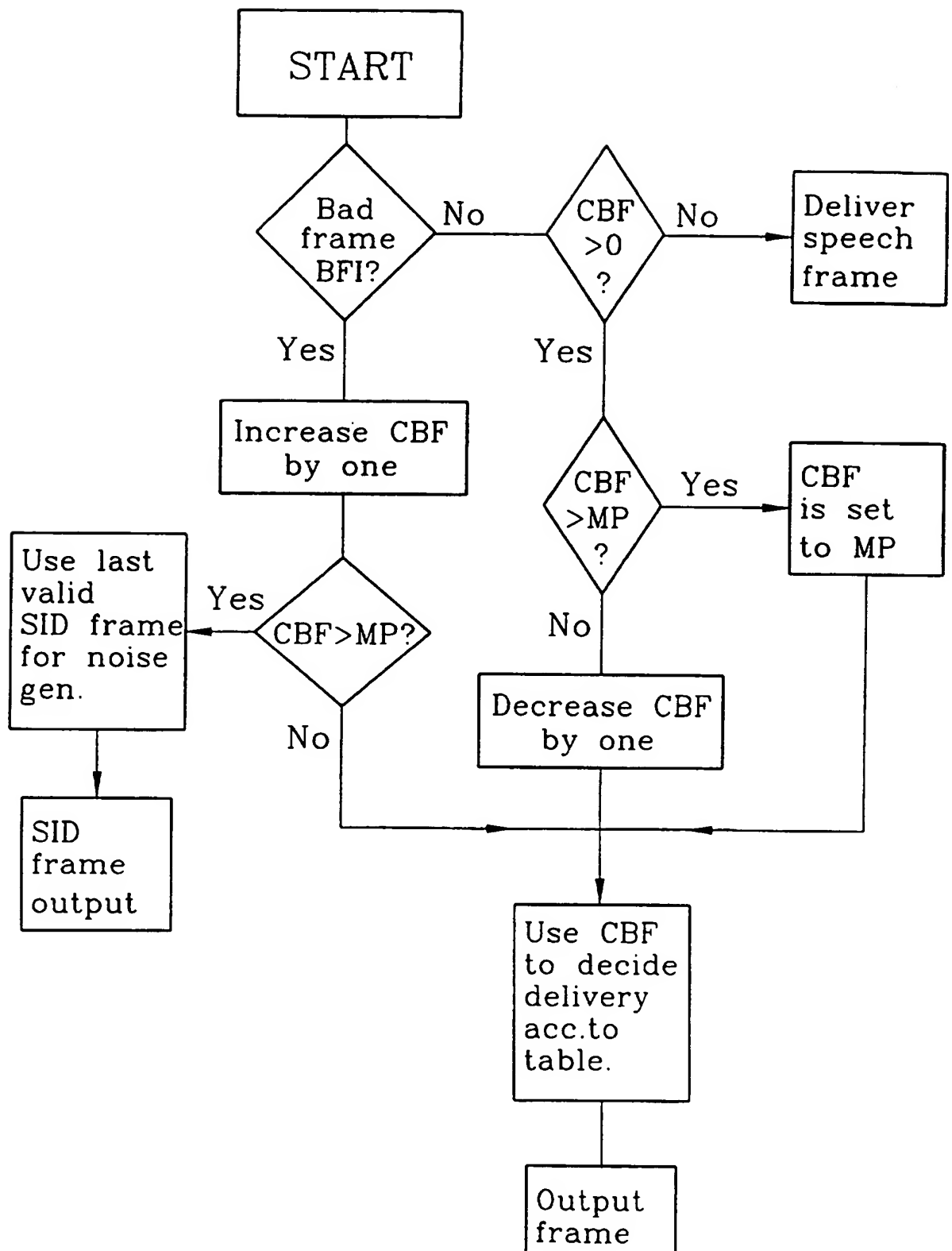
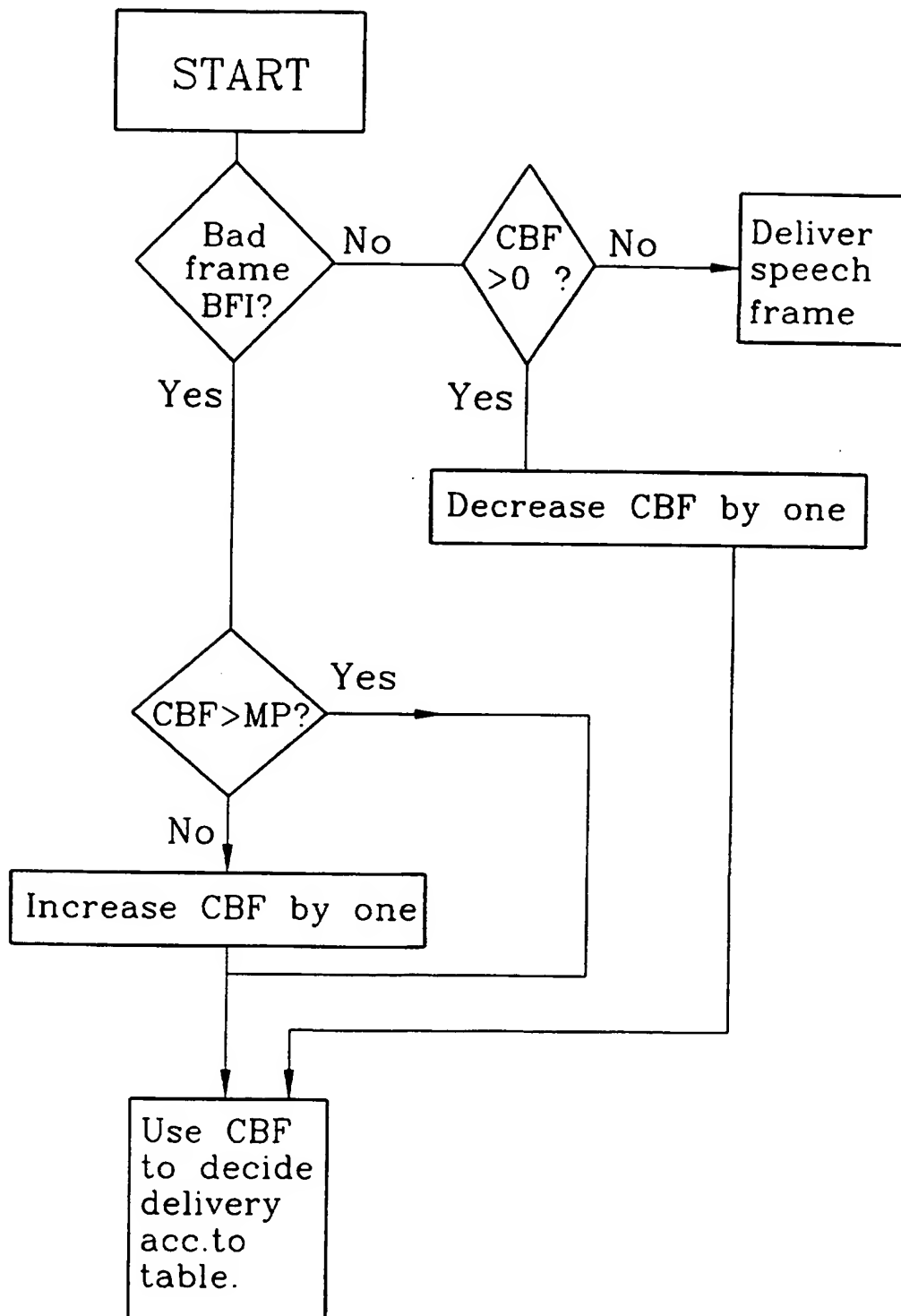


FIG. 8



## INTERNATIONAL SEARCH REPORT

International application No.

PCT/SE 96/00311

## A. CLASSIFICATION OF SUBJECT MATTER

IPC6: G10L 3/02, H04B 1/04

According to International Patent Classification (IPC) or to both national classification and IPC

## B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC6: G10L, H04B

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

SE,DK,FI,NO classes as above

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

WPI

## C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	GB 2256351 A (MOTOROLA INC.), 2 December 1992 (02.12.92), page 1, line 11 - line 35; page 7, line 9 - line 12	1,3-6,9-13, 15
Y	page 1, line 11 - line 35 --	2,7-8,14,16
X	European Telecommunication Standard, ETS 300 580-4, September 1994, "Comfort noise aspect for full rate speech traffic channels (GSM 06.12) , page 7, last paragraph, page 8, second and third paragraph, page 9, last paragraph --	1,3-5,9-13, 15
X	EP 0544101 A1 (NIPPON TELEGRAPH AND TELEPHONE CORPORATION), 2 June 1993 (02.06.93), abstract --	1,11,13,15

☒ Further documents are listed in the continuation of Box C.☒ See patent family annex.

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"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
10 June 1996	13 -06- 1996
Name and mailing address of the ISA/ Swedish Patent Office Box 5055, S-102 42 STOCKHOLM Facsimile No. +46 8 666 02 86	Authorized officer  Göran Magnusson Telephone No. +46 8 782 25 00

## INTERNATIONAL SEARCH REPORT

International application No.

PCT/SE 96/00311

## C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	EP 0599664 A2 (NEC CORPORATION), 1 June 1994 (01.06.94), abstract  --	1,11,13,15
Y	European Telecommunication Standard, ETS 300 580-3, September 1994, "Substitution and muting of lost frames for full rate speech channels (GSM 06.11), page 7, 2.1 and 2.2, page 8, 2.4  -- -----	2,7-8,14,16

**INTERNATIONAL SEARCH REPORT**  
Information on patent family members

01/04/96

International application No.  
PCT/SE 96/00311

Patent document cited in search report		Publication date	Patent family member(s)		Publication date
GB-A-	2256351	02/12/92	DE-A-	4216911	26/11/92
			FR-A,B-	2676876	27/11/92
EP-A1-	0544101	02/06/93	CA-A-	2081441	29/04/93
			JP-A-	5122165	18/05/93
EP-A2-	0599664	01/06/94	NONE		